REMARKS

Claims 1-3 are rejected under 35 U.S.C. 103(a) as being obvious over Morton (U.S. Patent Number 4,306,113). Claims 4-6 are rejected under 35 U.S.C. 103(a) as being unpatentable over Behrens, *et al.* (U.S. Patent Number 5,903,857) in view of Becker, *et al.* (U.S. Patent Number 5,929,628). In view of the following remarks, the rejections are respectfully traversed, and reconsideration of the rejections is requested.

In the present invention as claimed in claims 1 and 2, a filter characteristic measuring method includes generating an impulse signal, applying the impulse signal to a DUT having an analog filter through a digital channel, and, in response to the impulse signal, measuring a gain of the analog filter in the DUT and a frequency characteristic by using an output of the analog filter for testing an operation of the DUT in at least one of a test procedure and a product test.

In the present invention as claimed in claim 3, an analog filter characteristic method includes applying an impulse signal to an equalizing filter by using a digital channel of an automatic tester, and then obtaining an output response of the equalizing filterfilter and performing a differential and a fast Fourier transform (FFT) operation on the output response of the equalization filter so as to measure a boosting gain and a frequency response for testing an operation of the equalizing filter in at least one of a test procedure and a product test.

In the present invention as claimed in claims 4-6, a system for measuring a characteristic of a filter in a DUT employing an analog filter includes a digital channel that provides an impulse signal without applying a sine wave to the analog filter of the DUT and a digitizer for receiving an output signal of the analog filter in response to the impulse signal so as to measure the characteristic of the filter.

Morton discloses correcting errors in overall reproduction characteristics of an audio system installed in a residence. In a first step, a test signal is generated as an input to the audio system. The test signal may be a band-limited impulse function. Various impulse functions with various weightings and limited to various frequency bands can be used as the test signal to identify the characteristics of the audio system. A microphone detects the resulting sound output from the audio system in response to the test signal. The resulting sound output is stored on a storage medium. In the first step, the generation

of the band-limited impulse signal includes contents of a waveform PROM 808 which are outputted to a D/A converter 809 to generate a stepwise waveform similar to that shown in FIG. 9 of Morton. The higher harmonics of this waveform are removed by a D/A filter 810 to produce the desired test signal. In a second step, a process unit is used to read and process the stored data and to use this data to automatically adjust a specially designed equalizer to obtain the desired characteristics. In the second step, a test signal generator 2602 generates a signal comprising a predetermined number of impulse test waves (corresponding to the impulse signal used to originally test the audio system) for a period sufficient to enable a selected potentiometer drive motor to adjust its specific potentiometer of a time domain equalizer 2603. A D/A converter 2705 is driven to generate the required impulse test wave shape. This is filtered in a D/A filter 2706 to remove digitization noise. The impulse test waves generated by the test signal generator 2602 pass to the time domain equalizer 2603 and from the equalizer 2603 to a sample and hold circuit 2604. The impulse response from the equalizing filter is sampled and compared to an output of a D/A converter 2615 in order to adjust the potentiometers of the equalizer 2603, which affects the amplitude of the point on the impulse which the sample and hold circuit 2604 is presently sampling. In a third step the resulting adjusted equalizer is installed in the audio system so that the overall performance of the audio system is improved by correcting for errors inherent in the audio system.

Morton, *et al.* fails to teach or suggest a filter characteristic measuring method that includes generating an impulse signal and applying the impulse signal to a DUT having an analog filter through a digital channel and in response to the impulse signal, measuring a gain of the analog filter in the DUT and a frequency characteristic by using an output of the analog filter for testing an operation of the DUT in at least one of a test procedure and a product test, as claimed in claims 1 and 2. Instead, Morton specifically discloses applying an analog signal through an analog channel to the equalizer filter. Specifically, the D/A converter 2705 is driven to generate the required impulse test wave shape and the D/A filter 2706 removes digitization noise. The Examiner asserts that it would have been obvious to one of ordinary skill at the time the invention was made to teach the analog filter inside the DUT and applying the impulse signal to a DUT having an analog filter through a digital channel since the impulse test waves generated by the test signal generator 2601 pass to the equalizer 2603 and from the equalizer 2603 to a

sample and hold circuit 2604 which samples the amplitude of the resulting impulse at an instant during the occurrence of the impulse. The applicants respectfully disagree. In the present invention, applying an impulse signal to a DUT through a digital channel has the same meaning as simultaneously applying sine waves of all frequencies to the DUT. In Morton, applying the impulse test waves to the equalizer 2603 is not the same as applying sine waves of all frequencies, but is a specific sine wave which is used to adjust the potentiometers of the equalizer 2603. In addition, Morton specifically discloses that the digital signal is converted to an analog signal by D/A converter 2705 to generate a desired impulse test wave shape and the D/A filter 2706 removes digitization noise.

Further, Morton does not disclose measuring gain of the equalizer 2603 or a frequency characteristic in response to the impulse test waves by using an output of the equalizer 2603, as claimed in claims 1 and 2. Rather, in Morton, the characteristic of the audio system are measured in the first step in order to determine the adjustments to be made to the equalizer 2603. In Morton, the band-limited impulse function is applied to the audio system in order to determine the desired characteristics. Based on these characteristics, the equalizer 2603 of Morton is adjusted in response to the impulse test wave, so that the desired characteristics are achieved. The output of the equalizer 2603 is sampled and compared to a desired signal and then the potentiometers of the equalizer 2603 are adjusted accordingly. In Morton, there is no measurement of a gain or frequency characteristics of the equalizer 2603.

In addition, Morton fails to teach or suggest an analog filter characteristic method which includes applying an impulse signal to an equalizing filter by using a digital channel of an automatic tester, and then obtaining an output response of the equalizing filter and performing a differential and a fast Fourier transform (FFT) operation on the output response of the equalization filter so as to measure a boosting gain and a frequency response for testing an operation of the equalizing filter in at least one of a test procedure and a product test, as claimed in claim 3. In the present invention, applying an impulse signal to a DUT through a digital channel has the same meaning as simultaneously applying sine waves of all frequencies to the DUT. In Morton, applying the impulse test waves to the equalizer 2603 is not the same as applying sine waves of all frequencies, but is a specific sine wave which is used to adjust the potentiometers of the equalizer 2603. In addition, Morton specifically discloses that the digital signal is

converted to an analog signal by D/A converter 2705 to generate a desired impulse test wave shape and the D/A filter 2706 removes digitization noise. Further, Morton does not disclose measuring a boosting gain of the equalizer 2603 or a frequency response by using an output of the equalizer 2603, as claimed in claim 3. Rather, in Morton, the characteristic of the audio system are measured in the first step in order to determine the adjustments to be made to the equalizer 2603. In Morton, the band-limited impulse function is applied to the audio system in order to determine the desired characteristics. Based on these characteristics, the equalizer 2603 of Morton is adjusted in response to the impulse test wave, so that the desired characteristics are achieved. The output of the equalizer 2603 is sampled and compared to a desired signal and then the potentiometers of the equalizer 2603 are adjusted accordingly. In Morton, there is no measurement of a gain or a frequency response of the equalizer 2603.

Morton, *et al.* fails to teach or suggest the elements of the invention set forth in claims 1-3. Therefore, it is believed that the claims are allowable over the cited reference, and reconsideration of the rejections of claims 1-3 under U.S.C. 103(a) as being obvious over Morton, is respectfully requested.

With regard to the rejections of claims 4-6, Behrens, *et al.* discloses that when reading a recorded binary sequence from media, timing recovery 28 first locks to a write frequency by selecting, and input to a read channel, a write clock 54 through a multiplexor 60. Once locked to the write frequency, the multiplexor selects the signal 19 from the read head as the input to the read channel. A variable gain amplifier 22 adjusts the amplitude of the analog read signal 58, and an analog filter 20 provides initial equalization. A sampling device 24 samples an analog read signal 62 from the analog filter 20, and a discrete time equalizer filter 26 provides further equalization of the sample values 25 toward the desired response.

Behrens, *et al.* fails to teach or suggest a system for measuring a characteristic of a filter in a DUT employing an analog filter which includes a digital channel that provides an impulse signal without applying a sine wave to the analog filter of the DUT and a digitizer for receiving an output signal of the analog filter in response to the impulse signal so as to measure the characteristic of the filter, as claimed in claims 4-6. Instead, in Behrens, *et al.*, the variable gain amplifier 22 receives an analog read signal 58 and outputs and analog signal to multiplexor 60 which outputs an analog signal to the

analog filter 20. Therefore, the variable gain amplifier 22 does not provide an impulse signal without applying a sine wave to the analog filter 20, but rather applies a sine wave to the analog filter 20.

Becker, *et al.* is cited in the Office Action as disclosing a controller 206 for controlling the digital channel 212 and the digitizer 220. Becker, *et al.*, like Behrens, *et al.*, fails to teach or suggest a system for measuring a characteristic of a filter in a DUT employing an analog filter which includes a digital channel that provides an impulse signal without applying a sine wave to the analog filter of the DUT and a digitizer for receiving an output signal of the analog filter in response to the impulse signal so as to measure the characteristic of the filter, as claimed in claims 4-6.

Behrens, *et al.* and Becker, *et al.* fail to teach or suggest elements of the invention set forth in claims 4-6. Specifically, Behrens, *et al.* and Becker, *et al.* fail to teach or suggest a system for measuring a characteristic of a filter in a DUT employing an analog filter which includes a digital channel that provides an impulse signal without applying a sine wave to the analog filter of the DUT and a digitizer for receiving an output signal of the analog filter in response to the impulse signal so as to measure the characteristic of the filter, as claimed in claims 4-6. Accordingly, there is no combination of the references which would provide such teaching or suggestion. Neither of the references, taken alone or in combination, teaches or suggests the invention set forth in claims 4-6. Therefore, it is believed that claims 4-6 are allowable over the cited references, and reconsideration of the rejections of claims 4-6 under 35 U.S.C. § 103(a) based on Behrens, *et al.* and Becker, *et al.*, is respectfully requested.

In view of the foregoing remarks, it is believed that, upon entry of this Amendment, all claims pending in the application will be in condition for allowance. Therefore, it is requested that this Amendment be entered and that the case be allowed and passed to issue. If a telephone conference will expedite prosecution of the application, the Examiner is invited to telephone the undersigned.

Respectfully submitted,

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